Impact of channel estimation on MIMO transmission using sound waves

Luis Mendo Tomás
luis.mendo@upm.es
Dpto. de Señales, Sistemas y Radiocomunicaciones.
ETSI Telecomunicación. Universidad Politécnica de Madrid.
Av. Complutense, 30. 28040 Madrid.

***Abstract*—MIMO transmission relies on abstract mathematical techniques, which may be difficult to understand intuitively. Sound waves have proved to be a valuable tool for helping the students grasp the underlying concepts in MIMO and multipath propagation. This paper extends previous proposals in this area, and describes simple experiments that allow visual assessment of the impact of channel estimation on signal reception quality.**

I. Introduction

MIMO transmission is used by many modern wireless communication systems, such as LTE [1]. These techniques, and particularly spatial multiplexing, make use of abstract concepts from linear algebra [2]. In order to help the students understand them, it is desirable to carry out measurements that illustrate these concepts in an *intuitive* way. Since radio equipment is expensive, a good alternative is to use *sound waves*, which can be transmitted and received using laptop computers and standard audio equipment.

The fact that sound waves can be used for these purposes is based on the existing analogy between radio and sound propagation. In particular, sound waves are also subject to *multipath propagation*. This gives rise to time dispersion (or equivalently frequency selectivity [3]) and time-selective fading [4]; and also permits the use of MIMO techniques analogous to those used in radio [5]. With standard stereo sound outputs and inputs, the order of the MIMO system that can be set up is limited to 2×2. This is not a problem, because it suffices for illustrating the fundamental concepts.

Channel estimation is an important aspect that affects reception quality. Since the propagation channel is time-varying, *pilot symbols* need to be periodically inserted among the information symbols[1]. If the channel estimates are not sufficiently accurate, most of the information symbols will be demodulated incorrectly. Therefore, the energy of pilot symbols should be large enough so that the estimated channel coefficients are not too noisy, and their separation small enough so that the channel varies little between them.

This paper builds on the framework presented in [5] to illustrate the impact of channel estimation on the quality of the received signals, using 2×2 MIMO transmission with spatial multiplexing. Specifically, two aspects are observed:

- The effect of *channel variations* caused by movement of the receiver.
- The degradation caused by *clock rate differences* between transmitter and receiver.

[1]In OFMA there is an analogous requirement in the frequency domain.

The first of these aspects degrades the signal quality because the reference amplitude and phase obtained from the pilot symbols become less exact as time passes, which introduces a random phase shift and attenuation when demodulating the information symbols. The second causes a phase rotation at the receiver.

The purpose of this paper is to provide a means to *visualize* these effects, and thus make them more intuitive, using inexpensive equipment. This type of demonstrations have proved to be a valuable tool to help the students grasp the concepts that underpin wireless communication [4].

Section II briefly describes the used equipment. Section III discusses the measurement procedure, and Section IV presents example results. Finally, Section V gives the conclusions of the paper.

II. Required equipment

The equipment is the same as in [5]. The transmitter is a laptop computer with a pair of loudspeakers. The stereo transmitted signal is generated and reproduced using MATLAB. The receiver is a second computer with a pair of mono microphones connected to a stereo recording input by means of an adapter cable. The received stereo signal is recorded with an audio editor, saved to a file, and processed by the receiving program in MATLAB.

III. Measurement procedure

The proposed framework can accommodate the usual MIMO techniques, using either diversity (Alamouti; maximal ratio combining) or spatial multiplexing (V-BLAST; closed loop based on singular value decomposition of the channel matrix) [5]. Since the focus of this paper is the impact of channel estimation, only the V-BLAST mode is used, taken as a representative example. In this type of spatial multiplexing [2], the number of spatial streams coincides with the number of transmit and receive antennas, assumed to be equal. The transmitter pre-processing corresponds to the identity matrix, i.e. no pre-coding is applied; and the spatial post-processing matrix at the receiver is the inverse of the estimated channel matrix, assumed to be non-singular and sufficiently well conditioned.

The transmitted signal consists of time-domain raised-cosine pulses, given as $1-\cos(2\pi t/T_\mathrm{p})$, $0\leq t\leq T_\mathrm{p}$, modulated with BPSK or QPSK on a 320-Hz carrier. The pulse period T_p is selected as 1 second to avoid inter-symbol interference (the delay spread in a typical classroom may be up to several

TABLE I
MEASUREMENT CASES

Case	N	Modulation	Channel	Clock rate
1	32	BPSK	Static	Compensated
2	32	BPSK	Varying	Compensated
3	64	BPSK	Static	Not compensated
4	32	QPSK	Static	Compensated
5	32	QPSK	Varying	Compensated

tenths of a second). A sample frequency of 48000 samples/s is used.

The signal structure is as follows. A pilot symbol is first sent from the left loudspeaker, then from the right, and then two sequences (spatial layers) of a number N of data symbols are transmitted from both speakers using V-BLAST. Two cases are considered: binary bipolar sequences (BPSK) and quaternary IQ sequences (QPSK). The first case lends itself to a more intuitive interpretation of the post-processed signals at the receiver.

The receiver operations are similar to those applied in a real communications system. First a *bandpass filter* is applied to the received signals, in order to remove noise out of the signal band. This is important because mains electricity can introduce significant noise. The receiver then acquires *synchronization* and obtains an *estimate* $\hat{\mathbf{H}}$ of the channel matrix $\mathbf{H}$, using the two pilot pulses. The signal is later *converted to baseband*. This is done by multiplying by two quadrature sinusoids at the carrier frequency and applying a *lowpass filter* to remove the double-frequency components. Finally, the estimated channel matrix $\hat{\mathbf{H}}$ is inverted and the *spatial post-processing* is applied. The next step would be demodulating the signal of each spatial layer, but this is not done. Instead, the data part of the post-processed signals is displayed directly, to provide a more intuitive view of the quality of the received signals.

To *compensate for clock rate differences* between transmitter and receiver, the sample rate of the transmitted signal needs to be slightly adjusted [5]. Typical differences are of the order of tens of parts per million [4]. The required adjustment factor is obtained from a previous measurement using the same computers, and can be assumed to be constant across measurements.

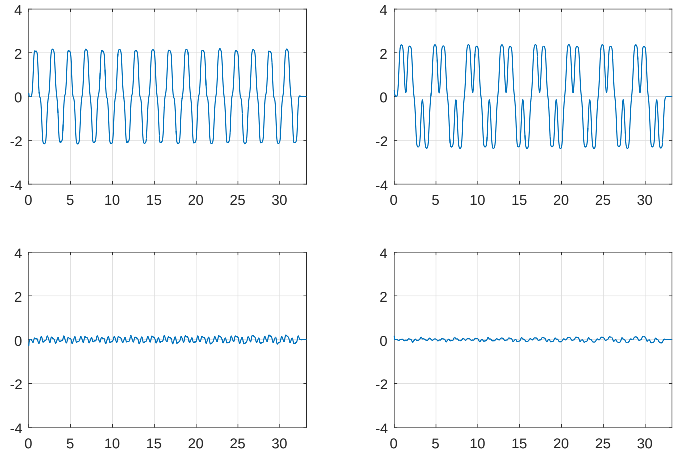
IV. MEASUREMENTS

The main purpose of the measurements is to observe the effect of outdated channel estimations. If the channel response gradually changes, data symbols that are further away from the initial pilot pulses will be more degraded than those located close to them. The effect can thus be illustrated by slowly moving at least one of the loudspeakers or microphones and observing the received, post-processed signals. An imperfect clock rate compensation will have a similar impact, as it is equivalent to a constant Doppler shift.

According to the above, several cases are considered, depending on:

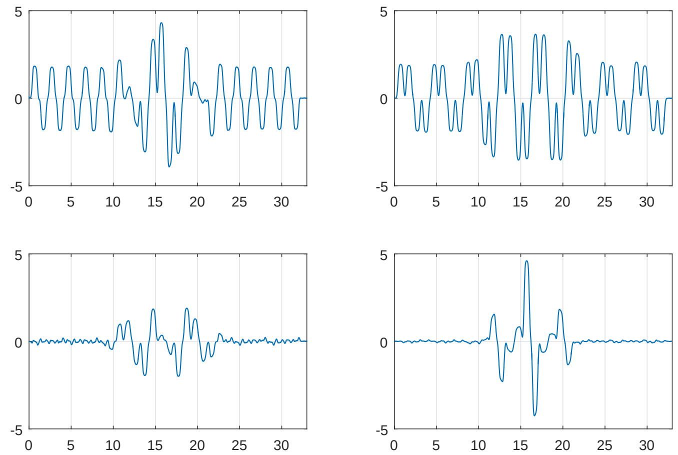
- The number N of data symbols.
- The modulation used: BPSK or QPSK. The first corresponds to a real, bipolar sequence for each spatial layer, whereas in the second each layer carries a complex (IQ) sequence.
- Whether the channel is static or time-varying. In the latter case, one of the microphones is moved slowly during part of the measurement.
- Whether clock rate compensation is applied or not. Lack of clock rate compensation produces a linear phase shift.

Table I summarizes the measurement cases.



Left/right: 1st/2nd spatial layers; up/down: I/Q branches.
Horizontal axis: time (s); vertical axis: normalized amplitude.

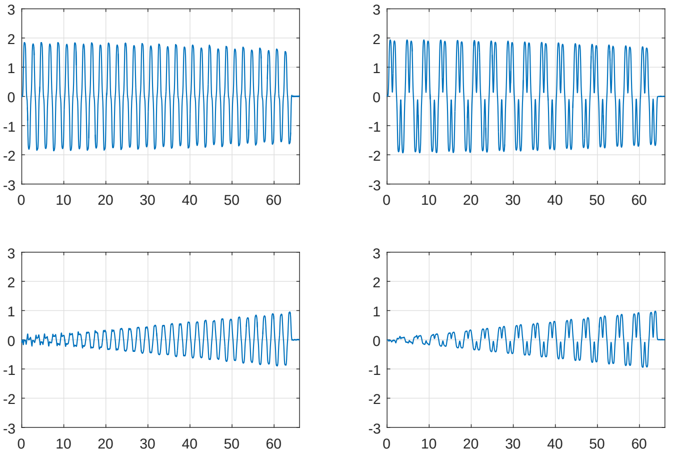
Fig. 1. BPSK, static channel



Left/right: 1st/2nd spatial layers; up/down: I/Q branches.
Horizontal axis: time (s); vertical axis: normalized amplitude.

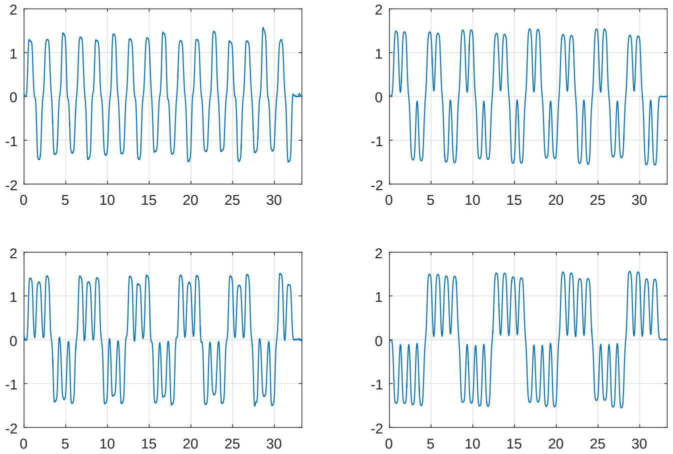
Fig. 2. BPSK, varying channel

The measurement results to be presented have been obtained in a small room, using standard loudspeakers and microphones. In Case 1 the microphones and all objects in the propagation environment are still during the measurement. This case is used as a reference. The first spatial layer transmits an alternating sequence of the form $+1, -1, +1, -1 \ldots$ whereas the second transmits $+1, +1, -1, -1, +1, +1 \ldots$ The results are seen in Figure 1, which shows the data section (i.e. without the two pilot symbols) of the post-processed received signals. The left and right parts in all figures correspond to the first and second spatial layers respectively, whereas up and down correspond to the I and Q branches. The horizontal axis is time in seconds, and the vertical axis is normalized



Left/right: 1st/2nd spatial layers; up/down: I/Q branches.
Horizontal axis: time (s); vertical axis: normalized amplitude.

Fig. 3. BPSK, static channel, no clock rate compensation



Left/right: 1st/2nd spatial layers; up/down: I/Q branches.
Horizontal axis: time (s); vertical axis: normalized amplitude.

Fig. 4. QPSK, static channel

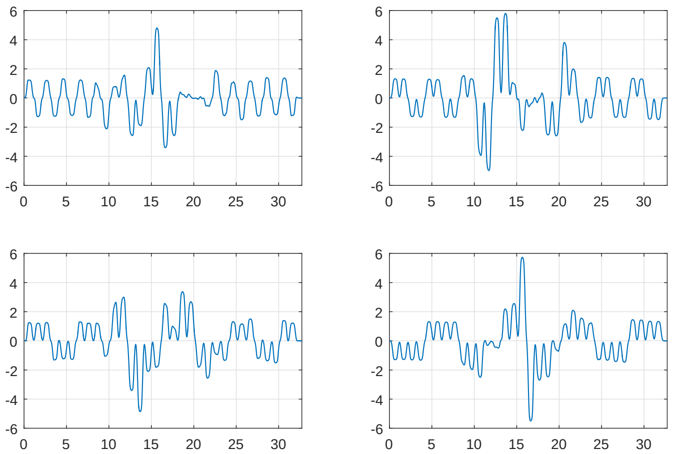
amplitude. The two sequences are clearly identifiable in the I branch, whereas the Q branch is approximately zero.

Case 2 differs from the preceding in that one of the microphones is moved. The movement starts approximately at data symbol 9 and stops at symbol 25; therefore it affects the central part of the time axis. The movement is done gradually, and spans roughly half a meter, i.e. half a wavelength ($\lambda = 340/320 = 1.06$ m). The final position of the microphone is approximately the same as the starting position. In Figure 2 it is seen that moving one of the microphones degrades the signal quality in both layers. This happens because each microphone is simultaneously receiving data from the two layers. The layers are then separated by the spatial post-processing using $\hat{\mathbf{H}}^{-1}$. Moving one microphone alters a single row of $\mathbf{H}$ or $\hat{\mathbf{H}}$, but this modifies all entries of $\hat{\mathbf{H}}^{-1}$.

The graphs in Figure 2 show how the central part of the signals is degraded by the fact that the channel estimation does not match the actual channel response. During the displacement of one of the receivers the signals become increasingly distorted. In particular, I-Q coupling is noticeable, specially in the lower half of the figures, in which the Q signals are no longer approximately zero. In the final part, where the receiver is brought back to its initial position, the transmitted sequences can again be recognized in the received signals, because the channel estimate becomes valid again.

Note how in the central part the transmitted sequences are not identifiable (specially in the first layer). This shows that the separation of the two spatial layers is not good, as a result of the channel estimation not reflecting the actual channel response. The figure thus illustrates that there is inter-layer coupling, not only I-Q coupling.

Case 3, shown in Figure 3, is a static channel like in Case 1, but no clock rate compensation is applied. As a result, the received phase slowly drifts. The first data symbols, located close to the pilot pulses, have a correct phase reference, but the reference gets gradually outdated as time progresses. This is seen in the figure as an increasingly large I-Q coupling. Part of the received signals, which should appear in the I branch, are seen in the Q branch. This happens in the two layers simultaneously, because the phase drift affects equally



Left/right: 1st/2nd spatial layers; up/down: I/Q branches.
Horizontal axis: time (s); vertical axis: normalized amplitude.

Fig. 5. QPSK, varying channel

to the four entries of the channel matrix $\mathbf{H}$. Note how the amplitude in the I branch decreases as the amplitude in the Q branch increases (in fact, the sum of squared amplitudes will be invariant, because the received signal power is constant).

Cases 4 and 5 are like 1 and 2, but using QPSK modulation. The I-branch sequences are the same as in the previous cases, whereas the Q branches use sequences $+1,+1,+1,-1,-1,-1\ldots$ for the first layer and $-1,-1,-1,-1,+1,+1,+1,+1\ldots$ for the second. These are clearly identified in Figure 4 (static channel).

When one of the receivers moves, Figure 5 again shows signal degradation because of the channel variations. The amount of displacement of the microphone is roughly the same as in Case 2. Here, as in Figure 2, the transmitted sequences cannot be recognized in the central part of the received signals. However, the degradation is more noticeable than in Case 2. This is due to the fact that here the Q branches are also active, and thus the coupling is more pronounced.

V. Conclusions

The proposed measurements use sound transmission to illustrate the degradation caused by outdated channel estimations. It is shown, in a very visual manner, how the received waveforms (after spatial post-processing) deviate from the ideal. The deviation is seen to increase as the microphone is moved further from its initial position, and disappears when the microphone is returned to its initial position. It is also observed how a difference between clock rates at the transmitter and the receiver causes a similar effect, in the form of a continuous phase rotation.

The measurements are simple and intuitive, and can be easily done as classroom demonstrations to reinforce the theoretical concepts in wireless communication courses.

Acknowlegement

The author thanks Prof. Pedro García del Pino for providing some of the audio hardware used in the measurements.

This work was supported by the Ministry of Economy and Competitiveness of Spain under Project TEC2014-57821-R.

References

[1] E. Dahlman, S. Parkvall, and J. Sköld, *4G: LTE/LTE-Advanced for Mobile Broadband*, 2nd ed. John Wiley and sons, 2014.

[2] D. Tse and P. Viswanath, *Fundamentals of Wireless Communication*. Cambridge University Press, 2005.

[3] L. Mendo, "Time-frequency characterization of a sound propagation channel as an educational tool," in *Simposium Nacional de la URSI*, Santiago de Compostela (Spain), 2013.

[4] L. Mendo, "Illustrating multipath propagation by means of sound waves," *IEEE Antennas and Propagation Magazine*, vol. 51, no. 4, pp. 173–184, August 2009.

[5] L. Mendo, "Illustrating MIMO transmission by means of sound waves," to appear in *IEEE Antennas and Propagation Magazine*, August 2016.